CLAIMS

I claim:

1. An analyzer system for tracking quality of voice services over a telephony network, said analyzer system connected to at least one node of at least one telephony network, said analyzer system comprising:

   a connector adapted for a non-intrusive, high-impedance interface to said telephony network;

   a high-impedance measurement subsystem capable of measuring quality of voice services passing through each said node on each said telephony network;

   a quality of service analysis subsystem that analyzes quality of service for each said telephony network for a plurality of calls made over said telephony network through said node; and

   a data store that records the analysis of the quality of service analysis subsystem.

2. The analyzer system of claim 1, further comprising a recording system for recording telephony communications.

3. The analyzer system of claim 1, further comprising a reporting subsystem that sends one or more said analyses of the quality of service analysis subsystem to a predetermined destination via the telephony network.

4. The analyzer system of claim 1, further comprising:

a monitoring subsystem that analyzes said recorded analyses and ascertains whether said quality of service falls below a predefined level; and

a notification subsystem whereby said analyzer system sends a message via the telephony network to a predefined destination when said quality of service falls below a predefined level.

5. An analyzer system for tracking quality of voice services over a telephony network and connected to at least two nodes corresponding to at least one telephony network, said analyzer system comprising:

at least one high-impedance measurement subsystem capable of measuring quality of service of each said telephony network by analyzing a voice communication passing through each said node on each said telephony network;

at least one quality of service analysis subsystem that analyzes quality of service for at least one of said telephony networks for the duration of at least one call made over said telephony networks and passing through said corresponding nodes connected to said telephony networks;

6. An analyzer system for tracking quality of voice services over a telephony network and connected to at least two nodes corresponding to at least one telephony network, said analyzer system comprising:

at least one additional telephony network; and

at least one switching subsystem corresponding to each quality of service analysis subsystem that, if during the course of a call said quality of service analysis subsystem detects a degradation of said quality of service on a first telephony network, said switching subsystem immediately switches the call to a second telephony network for uninterrupted servicing of said call.

7. The analyzer system of claim 5, further comprising at least one circuit-switched network wherein said analyzer system is connected to at least one node corresponding to said circuit-switched network and wherein said switching subsystem can select said circuit-switched network in lieu of said telephony networks and thereby switch said call to said circuit-switched network for uninterrupted servicing of said call.

8. A system for comparing quality of voice services in a plurality of telephony networks, said system comprising:

at least two telephony networks;

at least two analyzer systems wherein each such analyzer system is connected to a node on at least two of said telephony networks, each such analyzer system comprising a system capable of measuring quality of voice services over a telephony network;

at least one testing subsystem corresponding to one said analyzer system by which such analyzer system initiates at least two calls over at least two of said telephony networks to at least one other analyzer system and analyzes the quality of voice services between said first analyzer system and said second analyzer system over said telephony networks over which said calls were made;

at least one selection subsystem corresponding to each testing subsystem by which said corresponding analyzer system can select one telephony network from among the plurality of telephony networks tested for routing at least one call through said node corresponding to said corresponding analyzer system to another node, said selection based at least in part on an evaluation of said analyses by said corresponding testing subsystem.

9. The system of claim 8, wherein said system is a personal computer system comprising hardware and software components.

10. The analyzer system of claim 8, further comprising an circuit-switched network and wherein each said selection subsystem can select said circuit-switched network over the plurality of telephony networks for routing at least one call through said node corresponding to said corresponding analyzer system to another node, said selection based at least in part on an evaluation of said analyses by said corresponding testing subsystem.

11. A system for comparing quality of voice services in a plurality of telephony networks, said system comprising:

a first telephony network;

a second telephony network;

a first analyzer system connected to both a first node on said first telephony network and to a first node on said second telephony network, said first analyzer system comprising a system capable of measuring quality of voice services over a telephony network;

a second analyzer system connected to both a second node on said first telephony network and to a second node on said second telephony network, said second analyzer system comprising a system capable of measuring quality of voice services over a telephony network;

a first testing subsystem by which said first analyzer system initiates a first call over said first telephony network to said second analyzer system and analyzes the quality of voice services between said first analyzer system and said second analyzer system over said first telephony network, and by which said first analyzer system initiates a second call over said second telephony network to said

second analyzer system and analyzes the quality of voice services between said first analyzer system and said second analyzer system over said second telephony network;

a first selection subsystem by which said first analyzer system selects one such telephony network for at least one call routed through said first node to said second node, said selection based at least in part on an evaluation of said analyses by said first testing subsystem.

12. The system of claim 11, wherein said system is a personal computer system comprising hardware and software components.

13. The system of claim 11, further comprising:

a second testing subsystem by which said second analyzer system, upon receipt of said first call from said first analyzer system, analyzes the quality of voice services between said first analyzer system and said second analyzer system over said first telephony network, and by which said second analyzer system, upon receipt of said second call from said first analyzer system, analyzes the quality of voice services between said first analyzer system and said second analyzer system over said second telephony network;

a second selection subsystem by which said second analyzer system selects one such telephony network for a call routed through said second node to said first node, said selection based at least in part on an evaluation of said analyses by said second testing subsystem.

14. The system of claim 11, further comprising at least one additional analyzer system, each such additional analyzer system connected to an additional node on said first telephony network and to an additional node on said second telephony

network, each such additional analyzer system comprising a system capable of measuring quality of voice services over a telephony network, and wherein said first testing subsystem of said first analyzer system initiates a first call over said first telephony network to each said additional analyzer system and analyzes the quality of voice services between said first analyzer system and each said additional analyzer system over said first telephony network, and wherein said first analyzer system initiates a second call over said second telephony network to each said additional analyzer system and analyzes the quality of voice services between said first analyzer system and each said additional analyzer system over said second telephony network, and wherein said first selection subsystem of said first analyzer system selects one such telephony network from among those analyzed for at least one call routed through said first node to said second node and for at least one call routed to each of said additional nodes, each said selection based at least in part on an evaluation of said analyses by said first testing subsystem.

15. The system of claim 14, further comprising:

at least one additional testing subsystem by which at least one of said additional analyzer systems, upon receipt of said first call from said first analyzer system, analyzes the quality of voice services between said first analyzer and such said additional analyzer system over said first telephony network, and by which said second analyzer system, upon receipt of said second call from said first analyzer system, analyzes the quality of voice services between said first analyzer system and such said additional analyzer system over said second telephony network;

at least one additional selection subsystem corresponding to each said additional testing subsystem whereby each such said additional selection subsystem selects one such telephony network for a call routed through each such said additional node to said first node, said selection by each such said additional selection

subsystem based at least in part on an evaluation of said analyses by each said additional testing subsystem.

16. The system of claim 11, further comprising at least one additional telephony network wherein said first analyzer system is connected to a first node on said each additional telephony network, wherein said second analyzer system is connected to a second node on said each additional telephony network, wherein said first testing subsystem by which said first analyzer system initiates an additional call over said each additional telephony network to said second analyzer system and analyzes the quality of voice services between said first analyzer system and said second analyzer system over said each additional telephony network, and wherein said first selection subsystem selects a telephony network from a group of networks--said group of networks comprising said first telephony network, said second telephony networks, and said each additional telephony network--for at least one call routed through said first node to said second node, said selection based at least in part on an evaluation of said analyses by said first testing subsystem.

17. The system of claim 16 further comprising:

a second testing subsystem by which said second analyzer system, upon receipt of said first call from said first analyzer system, analyzes the quality of voice services between said first analyzer and said second analyzer over said first telephony network, and by which said second analyzer system, upon receipt of said second call from said first analyzer system, analyzes the quality of voice services between said first analyzer and said second analyzer over said second telephony network, and by which said second analyzer system, upon receipt of said each additional call from said first analyzer system, analyzes the quality of

voice services between said first analyzer and said second analyzer over said each additional telephony network.

a second selection subsystem by which said second analyzer system selects a telephony network from a group of networks--said group of networks comprising said first telephony network, said second telephony networks, and said each additional telephony network--for at least one call routed through said first node to said second node, said selection based at least in part on an evaluation of said analyses by said first testing subsystem.

18. The system of claim 16, further comprising at least one additional analyzer system, each such additional analyzer system connected to an additional node on said first telephony network and to an additional node on said second telephony network and an additional node on each additional telephony network, each such additional analyzer system comprising a system capable of measuring quality of voice services over a telephony network, and wherein said first testing subsystem of said first analyzer system initiates a first call over said first telephony network to each said additional analyzer system and analyzes the quality of voice services between said first analyzer system and each said additional analyzer system over said first telephony network, and wherein said first analyzer system initiates a second call over said second telephony network to each said additional analyzer system and analyzes the quality of voice services between said first analyzer system and each said additional analyzer system over said second telephony network, and wherein said first analyzer system initiates an additional call over each said additional telephony network to each said additional analyzer system and analyzes the quality of voice services between said first analyzer system and each said additional analyzer system over each said additional telephony network, and wherein said first selection subsystem of said first analyzer system selects, for each node tested, at least one such telephony network for each node from among those analyzed for at least one call routed through said first node to each of said

additional nodes, each said selection based at least in part on an evaluation of said analyses by said first testing subsystem.

19. The system of claim 18, wherein said system is a personal computer system comprising hardware and software components.

20. The system of claim 16, wherein at least one of said additional telephony networks is a hybrid VON/PSTN network.

21. The system of claim 11, wherein at least one of said telephony networks is a hybrid VON/PSTN network.

22. A system for comparing quality of voice services in a plurality of telephony networks, said system comprising:

a first telephony network;

a second telephony network;

a first analyzer system connected to both a first node on said first telephony network and to a first node on said second telephony network, said first analyzer system comprising a system capable of measuring quality of voice services over a telephony network;

a second analyzer system connected to both a second node on said first telephony network and to a second node on said second telephony network, said second analyzer system comprising a system capable of measuring quality of voice services over a telephony network;

a first testing subsystem by which both said first analyzer system initiates a first call over said first telephony network to said second analyzer system and analyzes

the quality of voice services between said first analyzer system and said second analyzer system over said first telephony network, and by which said first analyzer system, upon receipt of a second call from said second analyzer system, analyzes the quality of voice services between said first analyzer system and said second analyzer system over said second telephony network;

a second testing subsystem by which both said second analyzer system, upon receipt of said first call from said first analyzer system, analyzes the quality of voice services between said first analyzer system and said second analyzer system over said first telephony network, and by which said second analyzer system initiates a second call over said second telephony network to said first analyzer system and analyzes the quality of voice services between said first analyzer system and said second analyzer system over said second telephony network;

a first selection subsystem by which said first analyzer system selects at least one such telephony network for calls routed through said first node to said second node, said selection based at least in part on an evaluation of said analyses by said first testing subsystem; and

a second selection subsystem by which said second analyzer system selects at least one such telephony network for at least one call routed through said second node to said first node, said selection based at least in part on an evaluation of said analyses by said second testing subsystem.

23. The system of claim 22, wherein said system is a personal computer system comprising hardware and software components.

24. The system of claim 22, wherein at least one of said telephony networks is a hybrid VON/PSTN network.

25. A system for comparing quality of voice services on modern telephony networks, including VON, PSTN, and hybrid VON/PSTN networks, to an circuit-switched network such as PSTN, said system comprising:

a first telephony network;

a first circuit-switched network;

a first analyzer system connected to a first node on said first telephony network, said first analyzer system comprising a system capable of measuring quality of voice services over a telephony network;

a second analyzer system connected to a second node on said first telephony network, said second analyzer system comprising a system capable of measuring quality of voice services over a telephony network;

a first testing subsystem by which said first analyzer system initiates a first call over said first telephony network to said second analyzer system and analyzes the quality of voice services between said first analyzer system and said second analyzer system over said first telephony network;

a first selection subsystem by which said first analyzer system selects at least one such network for at least one call routed through said first node to said second node, said selection based at least in part on an evaluation of said analysis by said first testing subsystem.

26. The system of claim 25, wherein said system is a personal computer system comprising hardware and software components.

27. The system of claim 25, further comprising:

a second testing subsystem by which said second analyzer system, upon receipt of said first call from said first analyzer system, analyzes the quality of voice

services between said first analyzer and said second analyzer over said first telephony network; and

a second selection subsystem by which said second analyzer system selects at least one such network for a call routed through said second node to said first node, said selection based at least in part on an evaluation of said analysis by said second testing subsystem.

28. The system of claim 25, wherein said telephony network is a hybrid VON/PSTN network.

29. A system for comparing quality of voice services provided by a plurality of telephony service-providers, said system comprising:

at least two telephony service-providers;

at least two analyzer systems wherein each such analyzer system is connected to a node on a network for least two of said telephony service-providers, each such analyzer system comprising a system capable of measuring quality of voice services over a telephony service-provider network;

at least one testing subsystem corresponding to one said analyzer system by which such analyzer system initiates at least two calls through at least two of said telephony service-providers to at least one other analyzer system and analyzes the quality of voice services between said first analyzer system and said second analyzer system through said telephony service-providers with which said calls were made;

at least one selection subsystem corresponding to each testing subsystem by which said corresponding analyzer system can select one telephony service-provider from among the plurality of telephony service-providers tested for routing at least one call through said node corresponding to said corresponding

analyzer system to another node, said selection based at least in part on an evaluation of said analyses by said corresponding testing subsystem.

30. The system of claim 8, wherein said system is a personal computer system comprising hardware and software components.

31. The analyzer system of claim 8, further comprising an circuit-switched service-provider and wherein each said selection subsystem can select said circuit-switched service-provider in lieu of the plurality of telephony service-providers for routing at least one call through said node corresponding to said corresponding analyzer system to another node, said selection based at least in part on an evaluation of said analyses by said corresponding testing subsystem.

32. A system for comparing quality of voice services in a plurality of telephony service-providers, said system comprising:

a first telephony service-provider;

a second telephony service-provider;

a first analyzer system connected to both a first node on a network of said first telephony service-provider and to a first node on a network of said second telephony service-provider, said first analyzer system comprising a system capable of measuring quality of voice services over a telephony service-provider network;

a second analyzer system connected to both a second node on a network of said first telephony service-provider and to a second node on a network of said second telephony service-provider, said second analyzer system comprising a system capable of measuring quality of voice services over a telephony service-provider network;

a first testing subsystem by which said first analyzer system initiates a first call through said first telephony service-provider to said second analyzer system and analyzes the quality of voice services between said first analyzer system and said second analyzer system over said first telephony service-provider, and by which said first analyzer system initiates a second call through said second telephony service-provider to said second analyzer system and analyzes the quality of voice services between said first analyzer system and said second analyzer system over said second telephony service-provider;

a first selection subsystem by which said first analyzer system selects one such telephony service-provider for at least one call routed through said first node to said second node, said selection based at least in part on an evaluation of said analyses by said first testing subsystem.

33. The system of claim 11, wherein said system is a personal computer system comprising hardware and software components.

34. The system of claim 11, further comprising:

a second testing subsystem by which said second analyzer system, upon receipt of said first call from said first analyzer system, analyzes the quality of voice services between said first analyzer system and said second analyzer system through said first telephony service-provider, and by which said second analyzer system, upon receipt of said second call from said first analyzer system, analyzes the quality of voice services between said first analyzer system and said second analyzer system through said second telephony service-provider;

a second selection subsystem by which said second analyzer system selects one such telephony service-provider for a call routed through said second node to said first node, said selection based at least in part on an evaluation of said analyses by said second testing subsystem.

35. The system of claim 11, further comprising at least one additional analyzer system, each such additional analyzer system connected to an additional node on a network of said first telephony service-provider and to an additional node on a network of said second telephony service-provider, each such additional analyzer system comprising a system capable of measuring quality of voice services through a telephony service-provider, and wherein said first testing subsystem of said first analyzer system initiates a first call through said first telephony service-provider to each said additional analyzer system and analyzes the quality of voice services between said first analyzer system and each said additional analyzer system through said first telephony service-provider, and wherein said first analyzer system initiates a second call through said second telephony service-provider to each said additional analyzer system and analyzes the quality of voice services between said first analyzer system and each said additional analyzer system through said second telephony service-provider, and wherein said first selection subsystem of said first analyzer system selects one such telephony service-provider from among those analyzed for at least one call routed through said first node to said second node and for at least one call routed to each of said additional nodes, each said selection based at least in part on an evaluation of said analyses by said first testing subsystem.

36. The system of claim 14, further comprising:

at least one additional testing subsystem by which at least one of said additional analyzer systems, upon receipt of said first call from said first analyzer system, analyzes the quality of voice services between said first analyzer and such said additional analyzer system through said first telephony service-provider, and by which said second analyzer system, upon receipt of said second call from said first analyzer system, analyzes the quality of voice services between said first

analyzer system and such said additional analyzer system through said second telephony service-provider;

at least one additional selection subsystem corresponding to each said additional testing subsystem whereby each such said additional selection subsystem selects one such telephony service-provider for a call routed through each such said additional node to said first node, said selection by each such said additional selection subsystem based at least in part on an evaluation of said analyses by each said additional testing subsystem.

37. The system of claim 11, further comprising at least one additional telephony service-provider wherein said first analyzer system is connected to a first node on said each additional telephony service-provider, wherein said second analyzer system is connected to a second node on said each additional telephony service-provider, wherein said first testing subsystem by which said first analyzer system initiates an additional call through said each additional telephony service-provider to said second analyzer system and analyzes the quality of voice services between said first analyzer system and said second analyzer system through said each additional telephony service-provider, and wherein said first selection subsystem selects a telephony service-provider from a group of service-providers--said group of service-providers comprising said first telephony service-provider, said second telephony service-providers, and said each additional telephony service-provider--for at least one call routed through said first node to said second node, said selection based at least in part on an evaluation of said analyses by said first testing subsystem.

38. The system of claim 16 further comprising:

a second testing subsystem by which said second analyzer system, upon receipt of said first call from said first analyzer system, analyzes the quality of voice

services between said first analyzer and said second analyzer through said first telephony service-provider, and by which said second analyzer system, upon receipt of said second call from said first analyzer system, analyzes the quality of voice services between said first analyzer and said second analyzer through said second telephony service-provider, and by which said second analyzer system, upon receipt of said each additional call from said first analyzer system, analyzes the quality of voice services between said first analyzer and said second analyzer through said each additional telephony service-provider.

a second selection subsystem by which said second analyzer system selects a telephony service-provider from a group of service-providers--said group of service-providers comprising said first telephony service-provider, said second telephony service-providers, and said each additional telephony service-provider--for at least one call routed through said first node to said second node, said selection based at least in part on an evaluation of said analyses by said first testing subsystem.

39. The system of claim 16, further comprising at least one additional analyzer system, each such additional analyzer system connected to an additional node on said first telephony service-provider and to an additional node on said second telephony service-provider and an additional node on each additional telephony service-provider, each such additional analyzer system comprising a system capable of measuring quality of voice services through a telephony service-provider, and wherein said first testing subsystem of said first analyzer system initiates a first call through said first telephony service-provider to each said additional analyzer system and analyzes the quality of voice services between said first analyzer system and each said additional analyzer system through said first telephony service-provider, and wherein said first analyzer system initiates a second call through said second telephony service-provider to each said additional analyzer system and analyzes the quality of voice services between said first

analyzer system and each said additional analyzer system through said second telephony service-provider, and wherein said first analyzer system initiates an additional call through each said additional telephony service-provider to each said additional analyzer system and analyzes the quality of voice services between said first analyzer system and each said additional analyzer system through each said additional telephony service-provider, and wherein said first selection subsystem of said first analyzer system selects, for each node tested, at least one such telephony service-provider for each node from among those analyzed for at least one call routed through said first node to each of said additional nodes, each said selection based at least in part on an evaluation of said analyses by said first testing subsystem.

40. The system of claim 18, wherein said system is a personal computer system comprising hardware and software components.

41. The system of claim 16, wherein at least one of said additional telephony service-providers is a hybrid VON/PSTN service-provider.

42. The system of claim 11, wherein at least one of said telephony service-providers is a hybrid VON/PSTN service-provider.

43. A system for comparing quality of voice services in a plurality of telephony service-providers, said system comprising:

a first telephony service-provider;

a second telephony service-provider;

a first analyzer system connected to both a first node on a network of said first telephony service-provider and to a first node on said second telephony service-

provider, said first analyzer system comprising a system capable of measuring quality of voice services through a telephony service-provider;

a second analyzer system connected to both a second node on a network of said first telephony service-provider and to a second node on said second telephony service-provider, said second analyzer system comprising a system capable of measuring quality of voice services through a telephony service-provider;

a first testing subsystem by which both said first analyzer system initiates a first call through said first telephony service-provider to said second analyzer system and analyzes the quality of voice services between said first analyzer system and said second analyzer system through said first telephony service-provider, and by which said first analyzer system, upon receipt of a second call from said second analyzer system, analyzes the quality of voice services between said first analyzer system and said second analyzer system through said second telephony service-provider;

a second testing subsystem by which both said second analyzer system, upon receipt of said first call from said first analyzer system, analyzes the quality of voice services between said first analyzer system and said second analyzer system through said first telephony service-provider, and by which said second analyzer system initiates a second call through said second telephony service-provider to said first analyzer system and analyzes the quality of voice services between said first analyzer system and said second analyzer system through said second telephony service-provider;

a first selection subsystem by which said first analyzer system selects at least one such telephony service-provider for calls routed through said first node to said second node, said selection based at least in part on an evaluation of said analyses by said first testing subsystem; and

a second selection subsystem by which said second analyzer system selects at least one such telephony service-provider for at least one call routed through said

second node to said first node, said selection based at least in part on an evaluation of said analyses by said second testing subsystem.

44. The system of claim 22, wherein said system is a personal computer system comprising hardware and software components.

45. The system of claim 22, wherein at least one of said telephony service-providers is a hybrid VON/PSTN service-provider.

46. A system for comparing quality of voice services on modern telephony service-providers, including VON, PSTN, and hybrid VON/PSTN service-providers, to an circuit-switched service-provider such as PSTN, said system comprising:

a first telephony service-provider;

a first circuit-switched service-provider;

a first analyzer system connected to a first node on a network of said first telephony service-provider, said first analyzer system comprising a system capable of measuring quality of voice services through a telephony service-provider;

a second analyzer system connected to a second node on a network of said first telephony service-provider, said second analyzer system comprising a system capable of measuring quality of voice services through a telephony service-provider;

a first testing subsystem by which said first analyzer system initiates a first call through said first telephony service-provider to said second analyzer system and analyzes the quality of voice services between said first analyzer system and said second analyzer system through said first telephony service-provider;

a first selection subsystem by which said first analyzer system selects at least one such service-provider for at least one call routed through said first node to said second node, said selection based at least in part on an evaluation of said analysis by said first testing subsystem.

47. The system of claim 25, wherein said system is a personal computer system comprising hardware and software components.

48. The system of claim 25, further comprising:

a second testing subsystem by which said second analyzer system, upon receipt of said first call from said first analyzer system, analyzes the quality of voice services between said first analyzer and said second analyzer through said first telephony service-provider; and

a second selection subsystem by which said second analyzer system selects at least one such service-provider for a call routed through said second node to said first node, said selection based at least in part on an evaluation of said analysis by said second testing subsystem.

49. The system of claim 25, wherein said telephony service-provider is a hybrid VON/PSTN service-provider.

50. A system for measuring quality of voice services on modern telephony networks, including VON, PSTN, and hybrid VON/PSTN networks,, said system comprising an analyzer that measures quality of service of a signal transmitted over the telephony network.

51. The system of claim 50, further comprising a connector adapted to make a connection to said test point in the telephony network.

52. The system of claim 50, wherein said analyzer comprises a computer system.

53. The system of claim 52, wherein said computer system is a personal computer system comprising hardware and software components.

54. The system of claim 52, wherein said analyzer further comprises a configuration subsystem that configures said computer system to interface with said telephony network.

55. The system of claim 50, wherein said analyzer further comprises a quality of service analysis subsystem that performs an analysis of said quality of service of said signal.

56. The system of claim 55, wherein said analyzer further comprises a report generator that reports a result of said analysis of said quality of service of said signal.

57. The system of claim 50, wherein said analyzer further comprises a communications protocol analysis subsystem that performs an analysis of adherence to said communication protocol.

58. The system of claim 57, wherein said analyzer further comprises a report generator that reports a result of said analysis of adherence to said communication protocol.

59. The system of claim 50, wherein said analyzer further comprises:

a quality of service analysis subsystem that performs an analysis of said quality of service of said signal; and

a communications protocol analysis subsystem that performs an analysis of adherence to said communication protocol.

60. The system of claim 59, wherein said analyzer further comprises a report generator that reports both a result of said analysis of said quality of service of said signal and said analysis of adherence to said communication protocol.

61. The system of claim 51, wherein said analyzer comprises:

a computer system;

a configuration subsystem that configures said computer system to interface with said telephony network;

a quality of service analysis subsystem that performs an analysis of said quality of service of said signal; and

a report generator that reports both a result of said analysis of said quality of service of said signal and said analysis of adherence to said communication protocol.

62. The system of claim 61, wherein said system is a personal computer system comprising hardware and software components.

63. The system of claim 61, wherein said connector is a handset connector that couples to a telephone handset to make said connection.

64. The system of claim 61, wherein said connector is a base connector that couples to a telephone base to make said connection.

65. The system of claim 61, further comprising a recorder that records signals transmitted by the telephony network at said test point.

66. The system of claim 65, further comprising a reproduction unit that reproduces the recorded signal through the telephony network at said test point to permit a determination of a speech quality.

67. The system of claim 61, wherein said quality of service analysis subsystem comprises a subsystem that tests quality of service selected from a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, or a delay.

68. The system of claim 61, wherein said quality of service analysis subsystem comprises a subsystem that tests a telephony condition selected from an AC call-control signal, a DC call-control signal, a call progress tone, or a CLASS signal.

69. The system of claim 61, wherein said connector comprises a connector adapted to connect to a selected one of a telephone handset, a telephone base unit, a line

card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, or a router.

70. The system of claim 61, further comprising a comparison subsystem that compares results obtained from tests performed between two or more locations in said telephony network.

71. The system of claim 70, wherein said signals comprise signals selected from one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, or a delay.

72. The system of claim 70, wherein said signals comprise a signal obtained from a selected one of a telephone handset, a telephone base unit, a line card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, or a router.

73. The system of claim 70, wherein said signals comprise a signal obtained from a telephone handset and a signal obtained from a non-telephone device connected to a communication channel.

74. The system of claim 61, further comprising a recording subsystem that records a selected one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event.

75. The system of claim 74, wherein said recording subsystem comprises a non-volatile memory that maintains a record of a selected one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event.

76. The system of claim 61, further comprising a reproduction subsystem that reproduces from a record a selected one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event.

77. The system of claim 61, further comprising a synchronizing subsystem that synchronizes a first selected one of a waveform, a timing signal, or a line event with a second selected one of a waveform, a timing signal, or a line event.

78. The system of claim 61, wherein the computer system comprises a modeling subsystem that models mathematically a parameter of a programmable telephone equipment, said parameter selected from one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event, said modeling subsystem calculating an improvement in performance of said programmable telephone equipment in response to a change in a programmable characteristic of said telephone equipment.

79. The system of claim 61, further comprising:

a transmitter subsystem that sends over the telephony network information about a selected one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, a delay, a DC impedance, an AC impedance, a noise, an echo, a waveform, a line event, an AC call-control signal, a DC call-control signal, a call progress tone, or a CLASS signal.

80. The system of claim 79, further comprising a tone and waveform generator that sends tones and selected waveforms over the telephony network.

81. The system of claim 79, further comprising a synchronizing subsystem that synchronizes a first selected one of a tone, a waveform, a timing signal, or a line event with a second selected one of a tone, a waveform, a timing signal, or a line event.

82. The system of claim 79, further comprising a recorder that records speech and signals.

83. The system of claim 79, further comprising a digitizer that digitizes recorded speech and signals.

84. The system of claim 79, further comprising a scheduler that commands the transmitter to transmit said digitized recorded speech and signals as a file, said scheduler controlling and recording the time of transmission of said file so that a delay and a degradation of said signal in one-way transmission can be analyzed.

85. The system of claim 79, further comprising:

a recorder that records speech and signals;

a digitizer that digitizes recorded speech and signals; and

a scheduler that commands the transmitter to transmit said digitized recorded speech and signals as a file, said scheduler controlling and recording the time of transmission of said file, so that a delay and a degradation of said signal in one-way transmission can be analyzed.

86. The system of claim 79, further comprising:

a second transmitter that can generate a second call using a second network;

a second analyzer that measures quality of service of a signal transmitted over said second network, said second analyzer comprising:

a computer system;

a configuration subsystem that configures said computer system to interface with said second network;

an quality of service analysis subsystem that performs an analysis of said quality of service of said signal;

a report generator that reports a result of said analysis; and

a comparator that compares the relative performance of the telephony network and said second network.

87. The system of claim 86, wherein said second transmitter comprises a transmitter that repeatedly generates said test calls and said second analyzer comprises an analyzer that repeatedly analyzes said test calls to monitor during a telephone call whether a significant change in quality of service takes place.

88. The system of claim 61, further comprising a routing selector that selects a route based at least in part on said quality of service measurement.

89. The system of claim 61, further comprising:

a receiver subsystem that receives over the telephony network information about a selected one of a figure of merit for voice clarity, a figure of merit for voice

intelligibility, a voice level, a power, a timing signal, a delay, a DC impedance, an AC impedance, a noise, an echo, a waveform, a line event, an AC call-control signal, a DC call-control signal, a call progress tone and a CLASS signal.

90. The system of claim 89, further comprising a receiver that detects and recognizes tones and selected waveforms transmitted over the telephony network.

91. The system of claim 89, further comprising a synchronization detector that determines a synchronization between a first selected one of a tone, a waveform, a timing signal, or a line event with a second selected one of a tone, a waveform, a timing signal, or a line event.

92. The system of claim 89, further comprising a speech measurement subsystem that measures quality of service of a signal representative of speech.

93. The system of claim 89, further comprising a signal measurement subsystem that measures signal levels.

94. The system of claim 89, further comprising an analyzer that detects and analyzes DTMF signals.

95. The system of claim 89, further comprising a display subsystem that reports a selected result of one of quality of service measurement, signal level measurement, and signal type analysis.

96. The system of claim 89, further comprising:

a speech measurement subsystem that measures quality of service of a signal representative of speech;

a signal measurement subsystem that measures signal levels;

an analyzer that detects and analyzes DTMF signals; and

a display subsystem that reports a selected result of one of quality of service measurement, signal level measurement, and signal type analysis.

97. The system of claim 89, further comprising:

a second receiver that can accept a second call transmitted over a second network;

a second analyzer that measures quality of service of a signal transmitted over said second network

98. The system of claim 97, wherein said second analyzer comprises a comparator that compares the relative performance of the telephony network and said second network.

99. The system of claim 97 wherein said second receiver comprises an receiver that repeatedly receives said test calls and said second analyzer comprises an analyzer that repeatedly analyzes said test calls to monitor during a telephone call whether a significant change in quality of service takes place.

100. The system of claim 89, further comprising a routing selector that selects a route based at least in part on said quality of service measurement.

101. A system for analyzing adherence to a communication protocol, said system comprising:

a connector adapted to make a connection to a test point in the telephony network; and

an analyzer that analyzes adherence to a communication protocol at said test point, said analyzer comprising:

a computer system;

a configuration subsystem that configures said computer system to interface with said telephony network;

a communications protocol analysis subsystem that performs an analysis of adherence to said communication protocol; and

a report generator that reports a result of said communication protocol analysis.

102. The system of claim 101, wherein said system is a personal computer system comprising hardware and software components.

103. The system of claim 101, wherein said connector is a handset connector that couples to a telephone handset to make said connection.

104. The system of claim 101, wherein said connector is a base connector that couples to a telephone base to make said connection.

105. The system of claim 101, further comprising a recorder that records signals transmitted by the telephony network at said test point.

106. The system of claim 105, further comprising a reproduction unit that reproduces the recorded signal through the telephony network at said test point to permit an analysis of adherence to said communication protocol.

107. The system of claim 101, wherein said communication protocol analysis subsystem comprises a subsystem that tests an adherence to a communication protocol selected from an Internet protocol, a voice over Internet protocol, voice over frame relay, voice over asynchronous transfer mode, voice over digital subscriber line, and voice over cable.

108. The system of claim 101, wherein said connector comprises a connector adapted to connect to a selected one of a telephone handset, a telephone base unit, a line card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, or a router.

109. The system of claim 101, further comprising a comparison subsystem that compares adherence to a communication protocol as used in two or more locations in said telephony network.

110. The system of claim 109, wherein said communication protocol is selected from one of an Internet protocol, a voice over Internet protocol, voice over frame relay, voice over asynchronous transfer mode, voice over digital subscriber line, and voice over cable.

111. The system of claim 109, wherein said communication protocol is analyzed from a selected one of a telephone handset, a telephone base unit, a line card, an FXS

port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, or a router.

112. The system of claim 111, wherein said communication protocol is analyzed from a telephone handset and from a non-telephone device connected to a communication channel.

113. The system of claim 101, further comprising a recording subsystem that records the analysis of the communication protocol.

114. The system of claim 113, further comprising a non-volatile memory that maintains a record of the analysis of the communication protocol.

115. The system of claim 114, further comprising a reproduction subsystem that reproduces from a record the analysis of the communication protocol.

116. The system of claim 113, wherein the computer system comprises a modeling subsystem that models mathematically the communication protocol used by a programmable telephone equipment, said modeling subsystem calculating an improvement in performance of said programmable telephone equipment in response to a change in a programmable characteristic of said telephone equipment.

117. A system for measuring quality of voice services on modern telephony networks, including VON, PSTN, and hybrid VON/PSTN networks, and for analyzing adherence to a communication protocol, said system comprising an analyzer that

measures quality of service of a signal transmitted over the telephony network and analyzes adherence to a communication protocol at a test point in the telephony network.

118. The system of claim 117, further comprising a connector adapted to make a connection to said test point in the telephony network.

119. The system of claim 117, wherein said analyzer comprises a computer system.

120. The system of claim 119, wherein said system is a personal computer system comprising hardware and software components.

121. The system of claim 119, wherein said analyzer further comprises a configuration subsystem that configures said computer system to interface with said telephony network.

122. The system of claim 117, wherein said analyzer further comprises a quality of service analysis subsystem that performs an analysis of said quality of service of said signal.

123. The system of claim 122, wherein said analyzer further comprises a report generator that reports a result of said analysis of said quality of service of said signal.

124. The system of claim 117, wherein said analyzer further comprises a communications protocol analysis subsystem that performs an analysis of adherence to said communication protocol.

125. The system of claim 124, wherein said analyzer further comprises a report generator that reports a result of said analysis of adherence to said communication protocol.

126. The system of claim 122, wherein said analyzer further comprises:

a quality of service analysis subsystem that performs an analysis of said quality of service of said signal; and

a communications protocol analysis subsystem that performs an analysis of adherence to said communication protocol.

127. The system of claim 126, wherein said analyzer further comprises a report generator that reports both a result of said analysis of said quality of service of said signal and said analysis of adherence to said communication protocol.

128. The system of claim 117, wherein said analyzer comprises:

a computer system;

a configuration subsystem that configures said computer system to interface with said telephony network;

a quality of service analysis subsystem that performs an analysis of said quality of service of said signal;

a communications protocol analysis subsystem that performs an analysis of adherence to said communication protocol; and

a report generator that reports both a result of said analysis of said quality of service of said signal and said analysis of adherence to said communication protocol.

129. The system of claim 128, wherein said system is a personal computer system comprising hardware and software components.

130. The system of claim 128, wherein said connector is a handset connector that couples to a telephone handset to make said connection.

131. The system of claim 128, wherein said connector is a base connector that couples to a telephone base to make said connection.

132. The system of claim 128, further comprising:

a recorder that records signals transmitted by the telephony network at said test point.

133. The system of claim 132, further comprising a reproduction unit that reproduces the recorded signal through the telephony network at said test point to permit a determination of a speech quality.

134. The system of claim 128, wherein said quality of service analysis subsystem comprises a subsystem that tests quality of service selected from a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, or a delay.

135. The system of claim 128, wherein said quality of service analysis subsystem comprises a subsystem that tests a telephony condition selected from an AC call-control signal, a DC call-control signal, a call progress tone, or a CLASS signal.

136. The system of claim 128, wherein said communication protocol analysis subsystem comprises a subsystem that tests an adherence to a communication protocol selected from an Internet protocol, a voice over Internet protocol, voice over frame relay, voice over asynchronous transfer mode, voice over digital subscriber line, and voice over cable.

137. The system of claim 128, wherein said connector comprises a connector adapted to connect to a selected one of a telephone handset, a telephone base unit, a line card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, or a router.

138. The system of claim 128, further comprising a comparison subsystem that compares results obtained from tests performed between two or more locations in said telephony network.

139. The system of claim 138, wherein said signals comprise signals selected from one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, or a delay.

140. The system of claim 138, wherein said signals comprise a signal obtained from at least a selected one of a telephone handset, a telephone base unit, a line card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, or a router.

141. The system of claim 138, wherein said signals comprise a signal obtained from a telephone handset and at least a signal obtained from a non-telephone device connected to a communication channel.

142. The system of claim 128, further comprising a recording subsystem that records a selected one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event.

143. The system of claim 128, further comprising a non-volatile memory that maintains a record of a selected one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event.

144. The system of claim 128, further comprising a reproduction subsystem that reproduces from a record a selected one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event.

145. The system of claim 128, further comprising a synchronizing subsystem that synchronizes a first selected one of a waveform, a timing signal, or a line event with a second selected one of a waveform, a timing signal, or a line event.

146. The system of claim 128, wherein the computer system comprises a modeling subsystem that models mathematically a parameter of a programmable telephone equipment, said parameter selected from one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event, said modeling subsystem calculating an improvement in performance of said

programmable telephone equipment in response to a change in a programmable characteristic of said telephone equipment.

147. The system of claim 128, further comprising a tone and waveform generator that sends tones and selected waveforms over the telephony network.

148. The system of claim 128, further comprising a synchronizing subsystem that synchronizes a first selected one of a tone, a waveform, a timing signal, or a line event with a second selected one of a tone, a waveform, a timing signal, or a line event.

149. The system of claim 128, further comprising a recorder that records speech and signals.

150. The system of claim 128, further comprising a digitizer that digitizes recorded speech and signals.

151. The system of claim 128, further comprising a transmitter subsystem that sends over the telephony network information about a selected one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, a delay, a DC impedance, an AC impedance, a noise, an echo, a waveform, a line event, an AC call-control signal, a DC call-control signal, a call progress tone, or a CLASS signal.

152. The system of claim 151, further comprising a scheduler that commands the transmitter to transmit said digitized recorded speech and signals as a file, said

scheduler controlling and recording the time of transmission of said file, so that a delay and a degradation of said signal in one-way transmission can be analyzed.

153. The system of claim 151, further comprising:

a second transmitter that can generate a second call using a second network;

a second analyzer that measures quality of service of a signal transmitted over said second network and analyzes adherence to a communication protocol at said test point

154. The system of claim 153, wherein said second analyzer comprises a comparator that compares the relative performance of the telephony network and said second network.

155. The system of claim 154, further comprising a routing selector that selects a route based at least in part on said quality of service measurement.

156. The system of claim 155 further comprising a billing subsystem that records billing information pertaining to said route selected by said routing selector.

157. The system of claim 153, wherein said second transmitter comprises a transmitter that repeatedly generates said test calls and said second analyzer comprises an analyzer that repeatedly analyzes said test calls to monitor during a telephone call whether a significant change in quality of service takes place.

158. The system of claim 157, further comprising a rerouting selector that selects a new route based at least in part on whether a significant change in quality of service takes place.

159. The system of claim 158, further comprising a billing subsystem that records billing information pertaining to said new route selected.

160. The system of claim 128, further comprising a receiver subsystem that receives over the telephony network information about a selected one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, a delay, a DC impedance, an AC impedance, a noise, an echo, a waveform, a line event, an AC call-control signal, a DC call-control signal, a call progress tone and a CLASS signal.

161. The system of claim 160, further comprising a receiver that detects and recognizes tones and selected waveforms transmitted over the telephony network.

162. The system of claim 161, further comprising a synchronization detector that determines a synchronization between a first selected one of a tone, a waveform, a timing signal, or a line event with a second selected one of a tone, a waveform, a timing signal, or a line event.

163. The system of claim 160, further comprising a speech measurement subsystem that measures quality of service of a signal representative of speech.

164. The system of claim 160, further comprising a signal measurement subsystem that measures signal levels.

165. The system of claim 160, further comprising an analyzer that detects and analyzes DTMF signals.

166. The system of claim 160, further comprising a display subsystem that reports a selected result of one of quality of service measurement, signal level measurement, and signal type analysis.

167. The system of claim 160, further comprising:

a speech measurement subsystem that measures quality of service of a signal representative of speech;

a signal measurement subsystem that measures signal levels;

an analyzer that detects and analyzes DTMF signals; and

a display subsystem that reports a selected result of one of quality of service measurement, signal level measurement, and signal type analysis.

168. The system of claim 160, further comprising a second receiver that can accept a second call transmitted over a second network.

169. The system of claim 168, further comprising a second analyzer that measures quality of service of a signal transmitted over said second network and analyzes adherence to a communication protocol at said test point.

170. The system of claim 169, further comprising a comparator that compares the relative performance of the telephony network and said second network.

171. The system of claim 170 wherein said second receiver comprises an receiver that repeatedly receives said test calls and said second analyzer comprises an analyzer that repeatedly analyzes said test calls to monitor during a telephone call whether a significant change in quality of service takes place.

172. The system of claim 171, further comprising a rerouting selector that selects a new route based at least in part on whether a significant change in quality of service takes place.

173. The system of claim 160, further comprising a routing selector that selects a route based at least in part on said quality of service measurement.

174. The system of claim 128, further comprising a billing subsystem that records billing information pertaining to the communications network.

175. The system of claim 128, further comprising a routing selector that selects a route based at least in part on said quality of service measurement.

176. The system of claim 175 further comprising a billing subsystem that records billing information pertaining to said route selected by said routing selector.

177. The system of claim 128, further comprising a rerouting selector that selects a new route based at least in part on whether a significant change in quality of service takes place.

178. The system of claim 177, further comprising a billing subsystem that records billing information pertaining to said new route selected.

179. A method of testing quality of voice services from a source node to a destination node over a modern telephony network using at least two testers wherein a first tester is located at said source node and wherein a second tester is located at said destination node, said method comprising:

said first tester initiating a telephone call from the first tester to the second tester;

said first tester analyzing call progress tones in order to determine if the call is completed;

said first tester measuring the time from a dialing event by the first tester to an answer event by said second tester;

said first tester transmitting tones to said second tester for a purpose of enabling the second tester to receive said tones, analyze said tones, and determine the quality of transmission for said tones;

said first tester transmitting a plurality of reference wave files to said second tester for a purpose of enabling the second tester to receive said files, analyze said files, and determine the quality of transmission for said files;

said second tester transmitting tones to said first tester for a purpose of enabling the first tester to receive said tones, analyze said tones, and determine the quality of transmission for said tones;

said second tester transmitting a plurality of reference wave files to said first tester for a purpose of enabling the first tester to receive said files, analyze said files, and determine the quality of transmission for said files;

said first tester calculating a first score for the quality of said tones and said files said first tester received from the second tester;

said second tester calculating a second score for the quality of said tones and said files that said second tester received from said first tester;

said second tester sending to said first tester said second score and said first tester receiving from the second tester said score;

said first tester calculating an overall score for quality of voice service based at least in part on at least one of an attribute from a group of attributes comprising said first score, said second score, said time from said dialing event by the first tester to said answer event by said second tester; and

said first tester displaying said overall score on one of a peripheral group comprising a computer screen, a printer, an external communication device, a computer storage device, or a remote computer.

180. The method of claim 179, further comprising:

said first tester initiating a second telephone call on a second network from the first tester to the second tester;

said first tester analyzing said second call progress tones in order to determine if said second call call is completed;

said first tester measuring the time from a second dialing event by the first tester to a second answer event by said second tester;

said first tester transmitting tones over the second network to said second tester for a purpose of enabling the second tester to receive said tones, analyze said tones, and determine the quality of transmission for said tones;

said first tester transmitting a plurality of reference wave files to said second tester over the second network for a purpose of enabling the second tester to receive said files, analyze said files, and determine the quality of transmission for said files;

said second tester transmitting tones to said first tester over the second network for a purpose of enabling the first tester to receive said tones, analyze said tones, and determine the quality of transmission for said tones;

said second tester transmitting a plurality of reference wave files to said first tester over the second network for a purpose of enabling the first tester to receive said files, analyze said files, and determine the quality of transmission for said files;

said first tester calculating a third score for the quality of said tones and said files said first tester received from the second tester over the second network;

said second tester calculating a fourth score for the quality of said tones and said files that said second tester received from said first tester over the second network;

said second tester sending to said first tester said fourth score over the second network and said first tester receiving from the second tester said score;

said first tester calculating a second overall score for quality of voice service on said second network based at least in part on at least one of an attribute from a group of attributes comprising said third score, said fourth score, said time from said dialing event by the first tester to said answer event by said second tester over said second network; and

said first testing routing at least one subsequent call through said network or said second network based at least in part on a comparison of the overall score and the second overall score.

181. A method of testing quality of voice services over a digital telephone, said method comprising:

disconnecting the telephone base unit from the telephone handset, connecting said telephone handset to said tester, and connecting said tester to said telephone base via said telephone base unit's handset port;

said tester operating in a pass-through mode to enables normal operation between said telephone handset and said telephone base unit;

placing a test call via said telephone base unit;

said tester measuring signal levels of transmit signals and a receive signals at the handset port of said telephone base unit;

said tester determining the optimum signal parameters for simulating said telephone handset;

disconnecting said telephone handset from said tester;

said tester simulating the impedance of the telephone handset based on the measured parameters;

said tester generating test signals through the handset port of the telephone base unit;

said tester generating test wave files through the handset port of the telephone base unit;

said tester measuring signal said test signals and said files that are received through the telephone base unit; and

disconnecting the telephone base unit from said tester and reconnecting said telephone handset to the telephone base unit.

182. A method of using a communication tester to monitor quality of voice services as perceived by a caller, said method comprising:

interfacing said tester to a telephone line used by at least one caller without affecting the quality of the call or noticeably changing the level of transmitted signals and received signals;

monitoring dialed DTMF digits and storing them in a tester memory;

monitoring call progress tones and determining whether or not a call is connected;

measuring a time from an end of dialing to a first ringback.

recording a voice communication in a plurality of consecutive short-duration time divisions;

performing Digital Signal Processing to determine impairments such as echo, delay, crosstalk, or distortion;

performing speech recognition using DSP techniques to look for phrases that indicate difficulty with the call such "hello...hello...," "speak up...," or "I can't hear you..." as an indication of a problem with said call; and

generating a statistical log that includes at least one item of information such as a date and a time of said call, call duration, call quality for each measured sample period, or key words detected that indicate a problem; and

reporting statistics to a computer either in response to a request by such a computer or at pre-programmed times.

183. The method of claim 182, further comprising the element of monitoring said call for a sudden degradation in quality of service during said call and relaying information about any such degradation real-time to a service provider for corrective action.

184. The method of claim 182, further comprising the element of monitoring said call for a sudden degradation in quality during said call and relaying information

about any such degradation to a call routing system having the means to handoff said call to another carrier while said call remains in progress.

185. The method of claim 182, further comprising the element of monitoring said call for a DTMF string entered by a caller on said caller via a pushbutton keypad on a telephone indicating call quality is unsatisfactory and relaying information about any such DTMF string to a call routing system having the means to handoff said call to another carrier while said call remains in progress.

186. A system for measuring quality of voice services on modern telephony networks, including VON, PSTN, and hybrid VON/PSTN networks, and for analyzing adherence to a communication protocol, said system comprising:

a connector adapted to make a connection to a test point in the telephony network, said connector adapted to connect to at least a selected one of a telephone handset, a telephone base unit, a line card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, or a router.

an analyzer that measures quality of service of a signal transmitted over the telephony network and analyzes adherence to a communication protocol at said test point, said analyzer comprising:

a computer system;

a configuration subsystem that configures said computer system to interface with said telephony network;

a tone and waveform generator that sends tones and selected waveforms over the telephony network; and

a synchronizing subsystem that synchronizes a first selected one of a tone, a waveform, a timing signal, or a line event with a second selected one of a tone, a waveform, a timing signal, or a line event;

a transmitter subsystem that sends over the telephony network information about a selected one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, a delay, a DC impedance, an AC impedance, a noise, an echo, a waveform, a line event, an AC call-control signal, a DC call-control signal, a call progress tone, or a CLASS signal;

a scheduler that commands the transmitter subsystem to transmit said digitized recorded speech and signals as a file, said scheduler controlling and recording a time of transmission of said file so that a delay and a degradation of said signal in one-way transmission can be analyzed;

a receiver that detects and recognizes tones and selected waveforms transmitted over the telephony network and comprising a receiver subsystem that receives over the telephony network information about a selected one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, a delay, a DC impedance, an AC impedance, a noise, an echo, a waveform, a line event, an AC call-control signal, a DC call-control signal, a call progress tone, or a CLASS signal;

a synchronization detector that determines a synchronization between a first selected one of a tone, a waveform, a timing signal, or a line event with a second selected one of a tone, a waveform, a timing signal, or a line event;

a recorder that records speech and signals transmitted by the telephony network at said test point, said recorder comprising non-volatile memory;

a recording subsystem that records a selected one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event, said recording subsystem comprising a non-volatile memory that

maintains a record of at least one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event;

a reproduction unit that reproduces recorded signal through the telephony network at said test point to permit a determination of a speech quality, said reproduction unit further comprising a reproduction subsystem that reproduces from a record a selected one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event;

a modeling subsystem that models mathematically a parameter of a programmable telephone equipment, said parameter selected from one of a DC impedance, an AC impedance, a noise, an echo, a waveform, a timing signal, or a line event, said modeling subsystem calculating an improvement in performance of said programmable telephone equipment in response to a change in a programmable characteristic of said telephone equipment;

a quality of service analysis subsystem that performs an analysis of said quality of service of said signal, said quality of service analysis subsystem comprising: (a) a subsystem that tests quality of service selected from a figure of merit for voice clarity, voice intelligibility, a voice level, a power, a timing signal, or a delay; and (b) a subsystem that tests a telephony condition selected from an AC call-control signal, a DC call-control signal, a call progress tone, or a CLASS signal;

a communications protocol analysis subsystem that performs an analysis of adherence to said communication protocol, said communications protocol analysis subsystem comprising a subsystem that tests an adherence to a communication protocol selected from an Internet protocol, a voice over Internet protocol, voice over frame relay, voice over asynchronous transfer mode, voice over digital subscriber line, or voice over cable;

a speech measurement subsystem that measures quality of service of a signal representative of speech;

a signal measurement subsystem that measures signal levels;

a report generator that reports both a result of said analysis of said quality of service of said signal and said analysis of adherence to said communication protocol

a signal analyzer that detects and analyzes DTMF signals;

a display subsystem that reports a selected result of at least one from quality of service measurement, signal level measurement, or signal type analysis; and

a comparison subsystem that compares results obtained from tests performed between two or more locations in said telephony network, wherein: (a) said signals comprise signals selected from one of a figure of merit for voice clarity, a figure of merit for voice intelligibility, a voice level, a power, a timing signal, or a delay; (b) one of said signals comprise a signal obtained from a telephone handset; and (c) a second of said signals comprise a signal obtained from a selected one of a telephone base unit, a line card, an FXS port, and FXO port, an E&M port, a T1/E1/J1 digital trunk, an Ethernet port, audio in and out ports, a router, or from a non-telephone device connected to a communication channel;

a second analyzer that measures quality of service of a signal transmitted over said second network and analyzes adherence to a communication protocol at said test point wherein said second analyzer repeatedly analyzes said test calls to monitor during a telephone call whether a significant change in quality of service takes place, said second analyzer comprising:

a second transmitter that can generate a second call using a second network wherein said second transmitter comprises a transmitter that repeatedly generates said test calls; and

a second receiver that can accept a second call transmitted over a second network wherein said second receiver comprises a receiver that repeatedly receives said test calls;

a routing selector that selects a route based at least in part on said quality of service measurement;

a rerouting selector that selects a new route based at least in part on whether a significant change in quality of service takes place;

a billing subsystem that records billing information pertaining to at least one selected from a route selected by said routing selector, a reroute selected, or other carrier billing information;

a connector adapted for a non-intrusive, high-impedance interface to said telephony network;

a high-impedance measurement subsystem capable of measuring quality of voice services passing through each said node on each said telephony network;

a quality of service analysis subsystem that analyzes quality of service for each said telephony network for a plurality of calls made over said telephony network through each said node;

a data store that records the analysis of the quality of service analysis subsystem;

a reporting subsystem that sends one or more said analyses of the quality of service analysis subsystem to a predetermined destination via the telephony network;

a monitoring subsystem that analyzes said recorded analyses and ascertains whether said quality of service falls below a predefined level; and

a notification subsystem whereby said analyzer system sends a message via the telephony network to a predefined destination when said quality of service falls below a predefined level;

at least one high-impedance measurement subsystem capable of measuring quality of service of each said telephony network by analyzing a voice communication passing through each said node on each said telephony network;

at least one quality of service analysis subsystem that analyzes quality of service for at least one of said telephony networks for the duration of at least one call made over said telephony networks and passing through said corresponding nodes connected to said telephony networks;

at least one switching subsystem corresponding to each quality of service analysis subsystem that, if during the course of a call said quality of service analysis subsystem detects a degradation of said quality of service on a first telephony network, said switching subsystem immediately switches the call to a second telephony network for uninterrupted servicing of said call;

at least two analyzer systems wherein each such analyzer system is connected to a node on at least two of said telephony networks, each such analyzer system comprising a system capable of measuring quality of voice services over a telephony network;

at least one testing subsystem corresponding to one said analyzer system by which such analyzer system initiates at least two calls over at least two of said telephony networks to at least one other analyzer system and analyzes the quality of voice services between said first analyzer system and said second analyzer system over said telephony networks over which said calls were made;

at least one selection subsystem corresponding to each testing subsystem by which said corresponding analyzer system can select one telephony network from among the plurality of telephony networks tested for routing at least one call through said node corresponding to said corresponding analyzer system to another node, said selection based at least in part on an evaluation of said analyses by said corresponding testing subsystem; and

at least one circuit-switched network wherein said analyzer system is connected to at least one node corresponding to said circuit-switched network and wherein said system can select said circuit-switched network in lieu of said telephony networks.